Application No. 09/546,976 Amendment dated December 31, 2004 Response to Office Action of August 29, 2003 Atty. Docket No. 42390.P9093X Examiner William C. Schultz TC/A.U. 2664

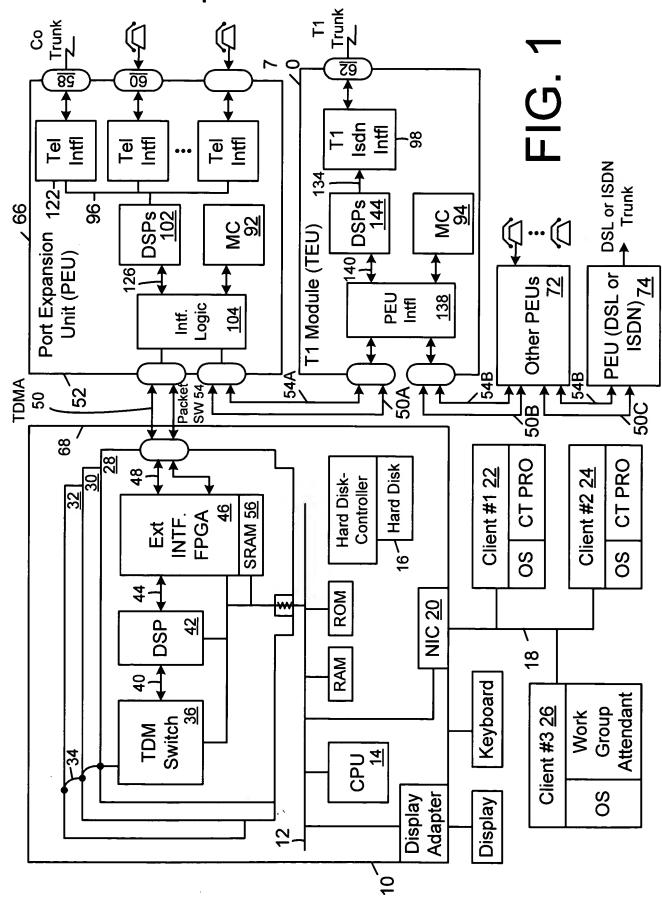
Amendments to the Figures

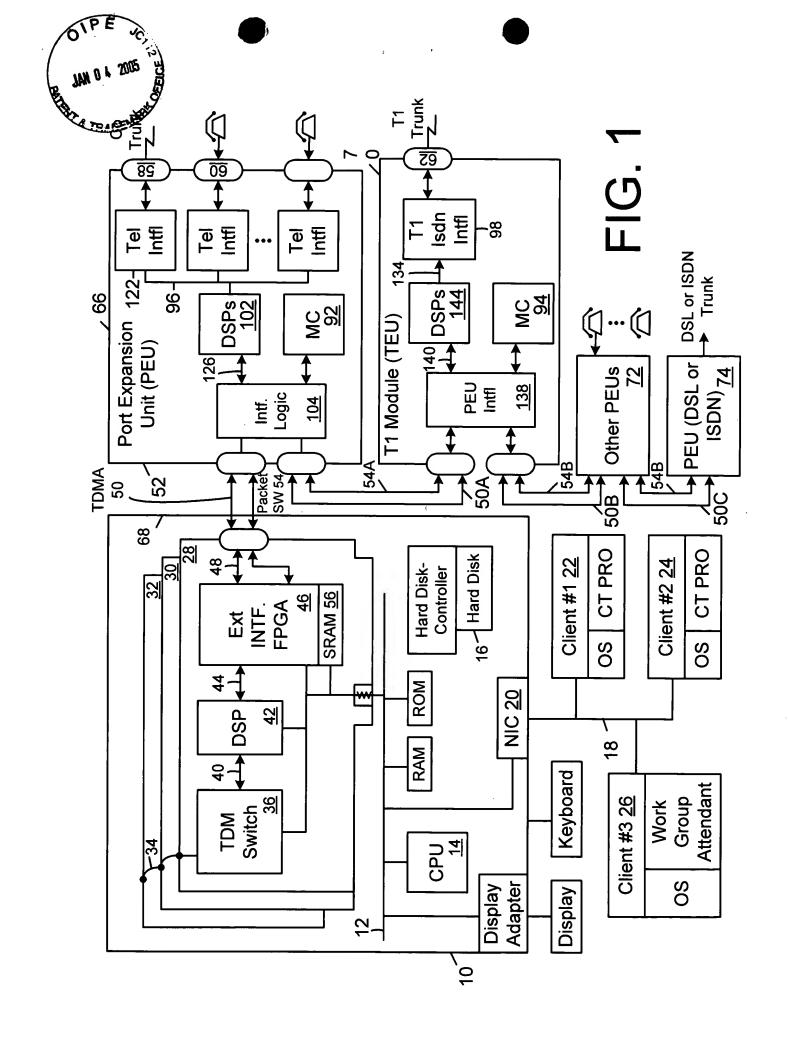
The attached sheet of figures includes changes to Figure 1. Duplicate element labeling has been corrected. Marked changes to Figure 1 and a full set of formal figures are attached.

Attachment: Annotated Sheet Showing Changes

Formal Figures

Replacement Sheet





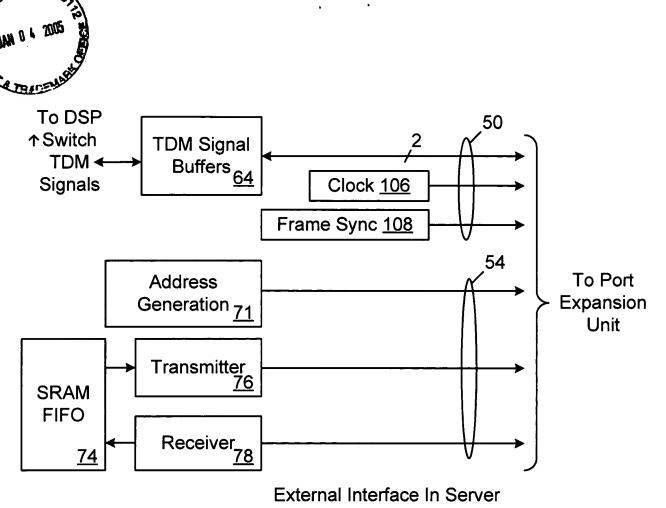


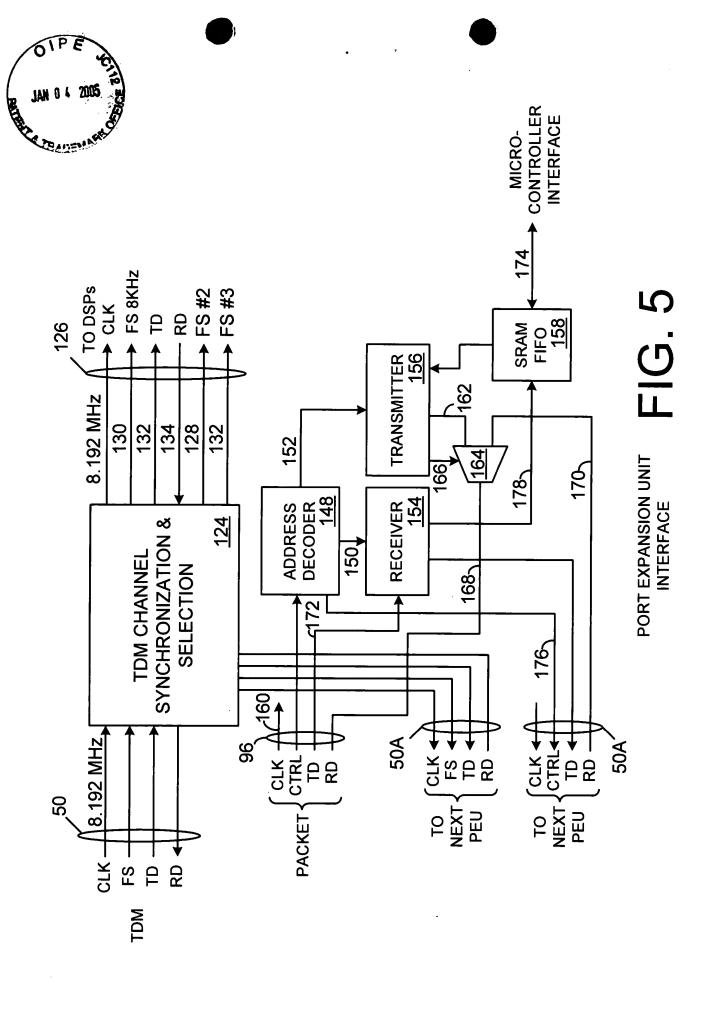
FIG. 2

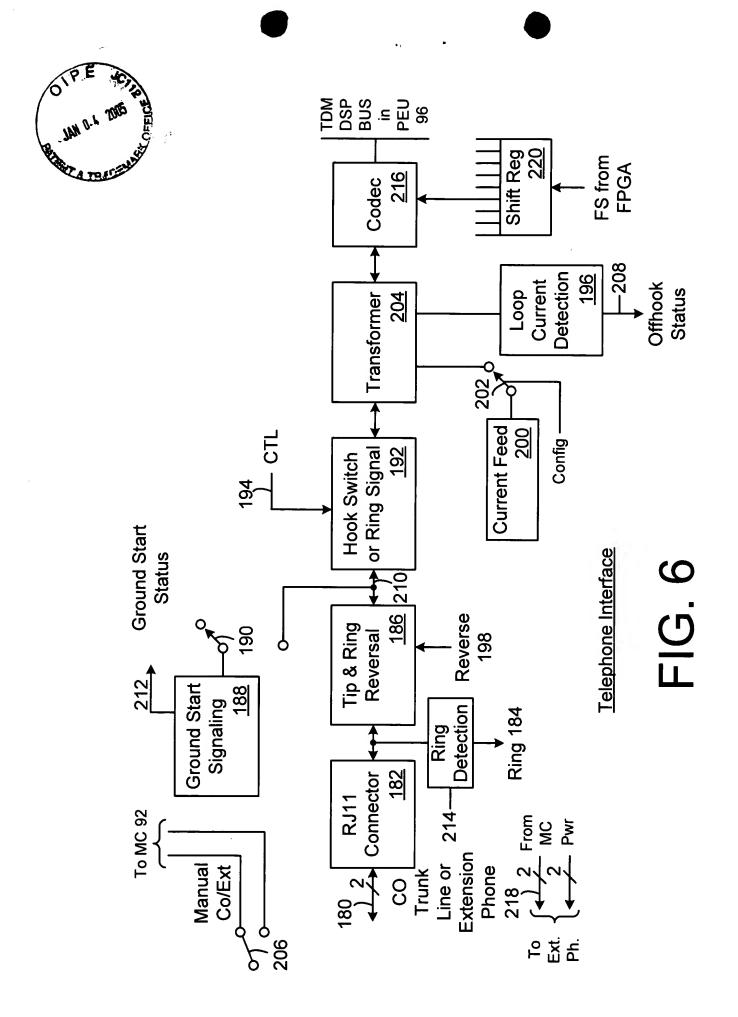
SIZE 80	DA 82	SA 84	TYPE 86	PAYLOAD 88	CRC 90
0.22 90	0, \ <u>v=</u>	0, (<u>v.</u>	· · · · <u>- oo</u>		0.10 00

FIG. 3

START 4-BIT I	MATCH BIT	STOP
---------------	--------------	------

FIG. 4





PE CONTRACTOR

PEU Processing

Microcontroller (hereafter MC) loads code for DSPs and programs the FPGA 222

DSP initializes each function and waits for channel assignment from MC 224

FPGA senses clock on bus 50 and learns its address by watching the match bit of the token 226

Once the address is learned, the FPGA interrupts the MC which records the address and sets a flag that it is now operational. MC sends a packet to PBX process on host that these PEU is now operational 228

MC starts generating a predefined hello packet that is sent to PBX indicating how many ports it has

FIG. 7A

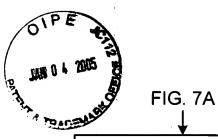
The PBX process receives the hello packet, assigns a channel on bus 50 to each port of the PEU and sends a packet back with that assignment 232

MC receives assignments and parses out assigned channels among all the DSPs on the PEU and gives the DSP a start processing signal 234

DSP buffers all 128 timeslots of data from bus 126 headed out to ports and buffers all 32 timeslots of data coming in from the ports 236

DSP also receives and buffers any tone data received from the MC or the port in a tone buffer for each port and receives the payloads of voice data from prerecorded messages and voicemail packets sent over the packet bus or received from the port in a voice buffer for each port 238

Read the timeslot data, voice data and tone data from each buffer of each port for each direction and weight data according to volume control weighting function for each type of data and sum the results 240



Write the resulting volume control weighted mixed data to the appropriate buffer 242

DTMF tone generation:
MC receives packet form
PBX host telling it which
DTMF tones to generate
on each channel
244

MC retrieves PCM data for desired tones and writes it into DSP tone buffer for the appropriate channel 246

DSP takes tone data and does volume control weighting on it and writes resulting sum data into output buffer for the channel 248

Play prerecorded announcements and voicemail data: receive control packet specifying a particular stream that is to be played on a particular port 250

PBX software broadcasts packets containing voice data to all peus over packet bus, each packet of a particular voicemail message or prerecorded message having a stream identifier with one packet being transmitted every

26 msec

252

FPGA at peu and copies any packets with correct PEU address or a broadcast PEU address that indicates all ports are supposed to process the message into the receive FIFO or MC 254

MC retrieves packets out of FIFO and looks at type field to determine if it is play data. If so, then the MC looks at the port address. If the port address is a broadcast address, then the MC looks at the stream ID and compares to stream ID previously identified in control messages for various ports to determine if there are any matches. MC writes voice data of payload section into input buffer of DSP that corresponds to any matches of 256 stream IDs

FIG. 7C

FIG. 7B



If the port address is not a broadcast address, then the MC writes the payload data from the packet into the input buffer of the DSP that corresponds to the port identified in the port address 258

DSP does volume control weighting and mixing of voice data with TDM data and tone data and writes summed data to output buffers of all ports on which message is to be played 260

DSP record data: data coming in from port is stored in TDM input buffer for each port. Data going out to port is stored in TDM output buffer. Volume control mix process reads TDM data traveling in each direction for port as well as packet data and tone data and weights each set of data in accord with weighting functions established by PBX software via control packets based upon user functionality requests 262

FIG. 7C

The summed volume control adjusted data is written into the record buffer of the DSP. Once 208 bytes have been accumulated, the DSP interrupts the MC. The MC reads the record data and packetizes it and stores it into transmit FIFO of FPGA. FPGA transmits packet over packet bus 264

DSP decodes caller ID data: MC detects first ring on any co port by polling the ring detector for all the ports and sets a status in the DSP to start transferring caller ID data. The DSP is constantly looking for caller ID data 266

DSP decodes incoming caller ID data, and when one byte has been received, interrupts the MC. The MC retrieves the caller ID data and verifies the checksum. Then the caller ID data is packetized and transmitted to the PBX software process. 268

The PBX process updates the attribute data of an object in memory created for this particular call with the caller ID data 270

FIG. 7D



FIG. 7C

DSP does DTMF detection:
DSP is constantly looking for
DTMF tones coming in from
every port. When detected,
DSP removes these DTMF
tones from the record buffer so
they will not be
recorded. 272

DSP interrupts MC when each tone starts and again when it ends and decodes each tone and silence between tones 274

MC retrieves each digit and packetizes it or a group therof and sends to PBX process. Silences between tones are also packetized and sent to PBX 276

DSP does MF detection: DSP interrupts MC when it stops. MC retrieves digits. When an entire sequence has been retrieved, MC packetizes sequence and sends it to the PBX software

DSP does call progress tone detection: DSP detects tones and builds a status word indicating which frequencies are active (1 bit per tone). Any time status changes, DSP interrupts MC

280

MC reads the status word each time it is interrupted and decodes the cadence to determine the call progress. The result is packetized and sent to the PBX process 282

DSP does eco cancellation: DSP removes echoes of DTMF prompt tones form any record data using adaptive filters implemented in software and known algorithms 284

DSP does AGC to normalize variations between line quality: known algorithms are used to do this 286

Done, return to start

288



MC - Telephone Interface Processing for Incoming Call

MC receives configuration data from PBX at startup and send command to did switch 202 of each port to set for a co port or extension phone as appropriate 287

MC reads ground start/loop start configuration status at startup and sends a command to switch 190 to set it for the proper signaling protocol 289

MC polls loop current detector every 10 msec to read status. IF get 2-3 status which are the same, then that is considered to be the new status and is reported by packet to PBX 290

MC polls ring detector every 2 msec. Status is debounced and is then reported by packet to PBX 292

FIG. 8A

MC communicates digital data with extension phones: data to be sent to phone is gathered from packets sent by PBX. MC polls extension phones for data to be sent back from phone by addressing each one individually in turn. Extension phone responds to poll by sending any data it has or an "ack, no data" message. Any data to be sent to phone is sent during times when phone is not being polled. 294

If the phone which was polled does not respond for N consecutive polls, the MC marks pcomm for that phone as down. Another process retries the downed ports every second to determine if they have come back up yet. The table of which ports have pcomm status up or down is reported by packets to the PBX process 296

Port configuration process periodically reads manual switch 206 of each port to determine if there has been any configuration change 298



FIG. 8A

If any configuration change has happened, a packet is sent to the PBX process signaling the changed configuration 300

Monitor for incoming did call on all co ports configured for did:
MC examines status of loop current detection and ring status and draws conclusion as to whether a did incoming call is ready to be sent by co 302

When the MC detects loop current and no ring signal, it sends a command to the tip and ring reversal circuit to cause it to flip the tip line voltage polarity to +48 volts relative to ring to signal the co that the PBX was ready to receive the dialed digits. The MC then sends a control packet to the PBX indicating a did call is inbound 304

Co sends dialed digits as DTMF tones. DSP detects DTMF tones and interrupts MC. MC reads digits and packetizes and sends to PBX process₃₀₆

FIG. 8B

The PBX process reads the dialed digits of the did call and looks up the extension that is mapped to that dialed number and sends a command packet to the appropriate PEU saying "ring port x" 308

MC activates line 194 to ring signal circuit 192 to cause it to start sending ring signal to extension named in control packet at cadence given in control packet 310

When the polling of the loop current detector indicates that the extension phone has gone offhook, MC sends a control packet with that change in status to the PBX process 312

PBX sends control packet back to MC saying answer call and sets up the switch connection between the co port and the extension port 314

MC responds to control packet
by deactivating reverse
command on line 198 to cause
tip and ring reversal circuit to set
tip and ring polarities back to
normal to signal co that
extension phone has answered
call 316

FIG. 8C



FIG. 8B

CO stops sending ringback tone to caller and connects analog signals from caller to PBX

318

PEU receives digitized data from CO and routes to switch card over TDM Bus timeslot assigned to CO port where gets sent back to this PEU or another PEU on the timeslot assigned to the extension called 320

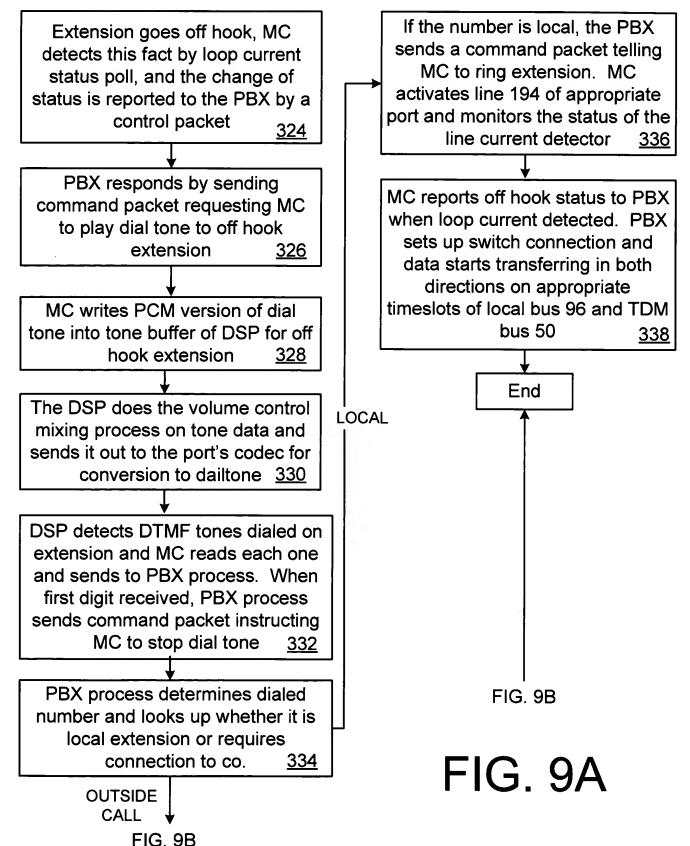
DSP receives data from bus 50 timeslot assigned to called extension and puts in on the timeslot of the local bus 96 assigned to the port of the called extension. Vice Versa for data coming in from the codec of the port of the called ext. 322

FIG. 9A

FIG. 8C



MC - Telephone Interface Processing for Incoming Call





OUTSIDE CALL

PBX process optionally does least cost routing and optionally does toll restrictions. Then sends command packet to MC of a PEU with an available port connected to cotelling it to go off hook 340

MC activates line 194 to go off hock on co port and monitors loop current detector 196 to determine when co goes off hock. When the co goes off hook, this status change is reported to the PBX 342

The co responds with dial tone and the DSP detects it and interrupts MC. MC reports dial tone to PBX in control packet 344

PBX process responds to dial tone by sending a control packet to MC with dial string 346

MC responds by writing PCM data for dial string characters into appropriate tone buffer of DSP assigned to co port 348

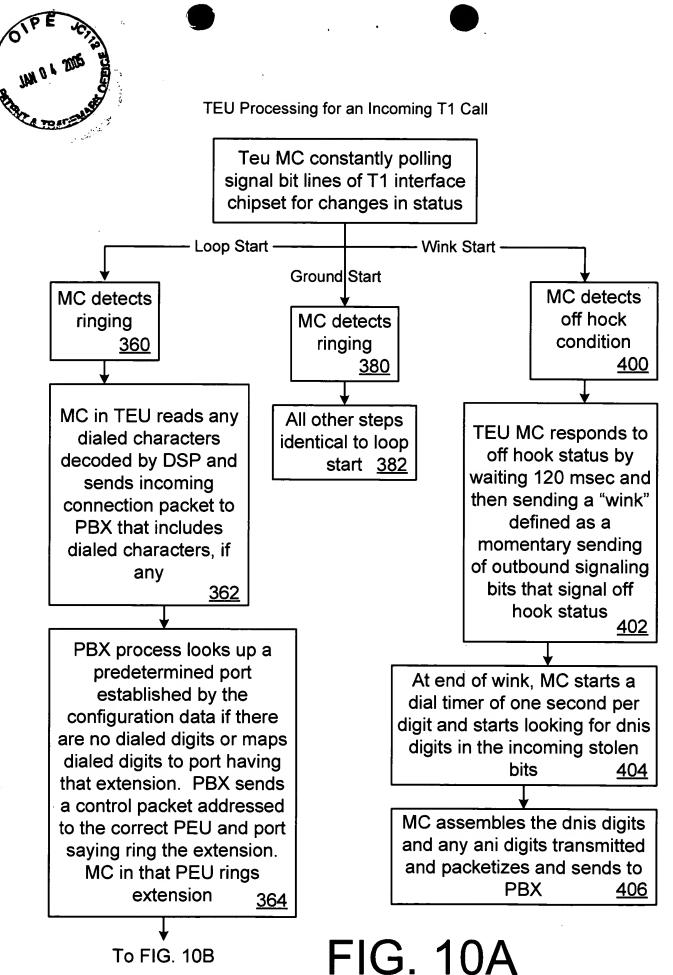
DSP does volume control mixing and sends resulting mixed, volume adjusted tone and other data to codec on appropriate timeslot for conversion to analog signal 350

FIG. 9A

Co sends ringback tone to PEU port. DSP detects ringback tone and interrupts MC. MC interprets and reports ringback status to PBX process. Ringback and any conversation after dialed phone picked up or any voicemail prompts are heard by extension 354

MC reports dial completed to PBX. PBX responds by making switch connections 352

FIG. 9B





From FIG. 10A

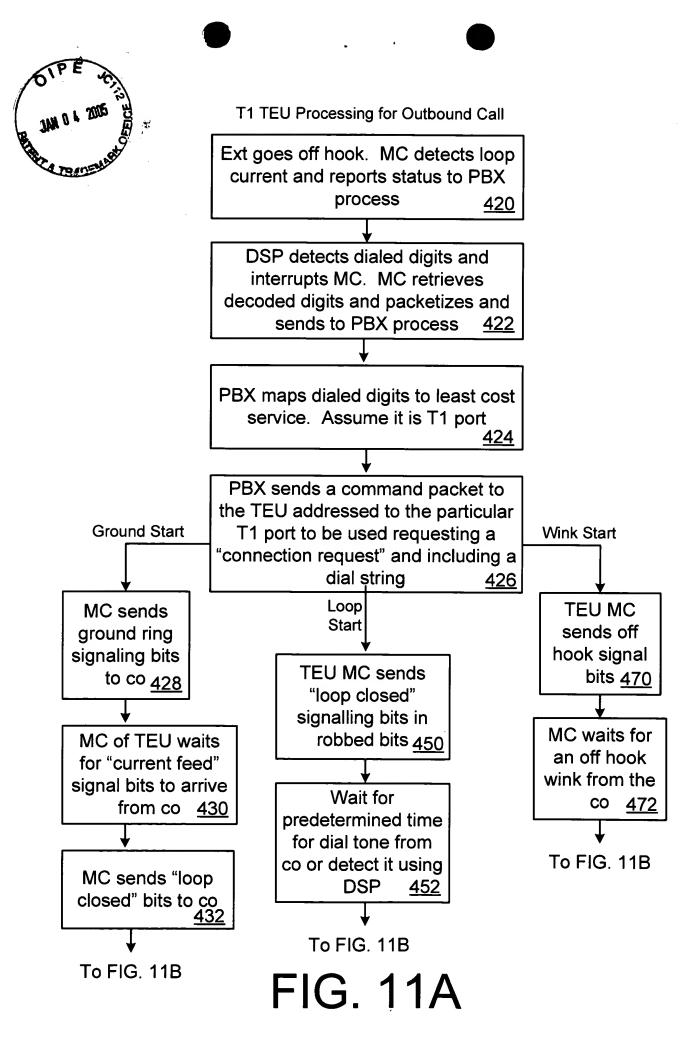
Extension phone goes off hook and this status is detected by MC and reported to PBX process

366

PBX process sends back a "connection response" and writes bits to switch card to set up the connection from the T1 port and extension port 368

TEU answers call by writing control bits into outbound lines of T! interface chipset which puts them into stolen bit positions of outbound T1 line

FIG. 10B



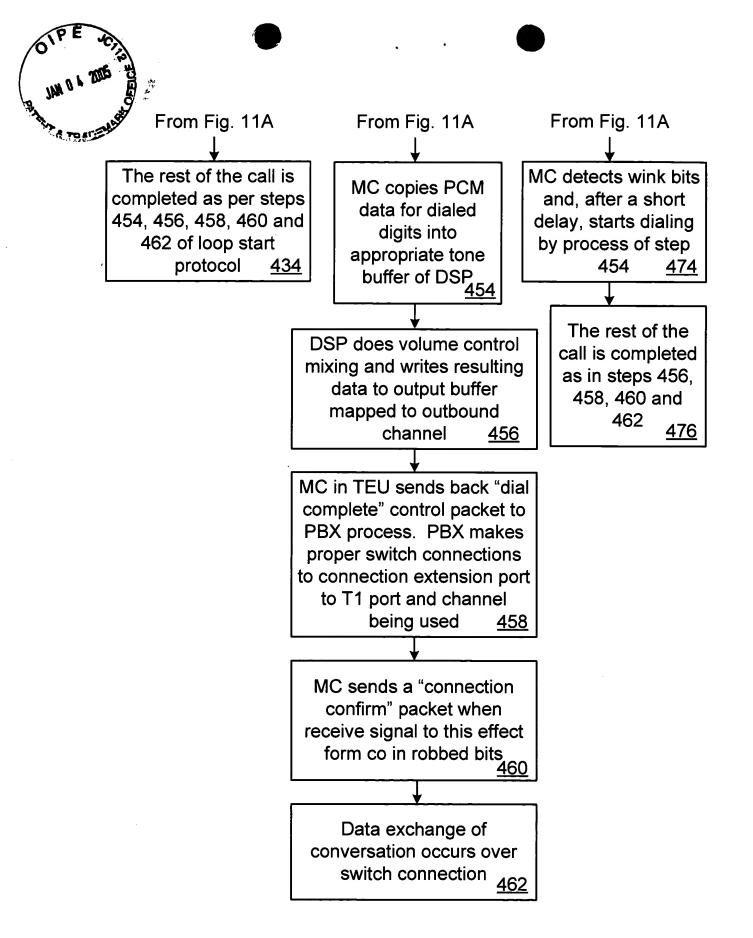


FIG. 11B



Switch Card DSP Conferencing Process

First DSP does known echo canceling algorithm on output of first 13 of 32 TDM channels devoted to conferencing

472

Second DSP does known echo canceling routine on outpt of next 13 of 32 conferencing channels and passes first 13 and last 6 on unchanged to the third DSP 474

Third DSP does echo canceling on last 6 channels and then does automatic gain control function on all 32 conference channels in each direction so each participant hears the others at the same approximate volume 476

DSP 3 uses conference setup data written into its memory by PBX process to determine which channels belong to which conferences 478

FIG. 12

For each conference, DSP 3
does volume control by
calculating the sum of all data in
the timeslots assigned to the
conference plus any beep tone
used to alert conferees to joiner
by a new conferee 480

DSP 3 calculates the input data for each conferee's channel so each conferee hears all the other participants but does not hear his or her own voice. The input for each channel is the conference sum minus the output data of each channel

The conference channel input data for each conferee's channel calculated in step 482 is multiplied by the conference volume

DSP 3 sends the calculation results for each channel as the input data for that channel. All other input data from the PEU for other channels is passed by DSP 3 unchanged. DSP 2 and DSP 1 pass all data through unchanged 486



Analog Loop Start Protocol Process - CO Originates Call

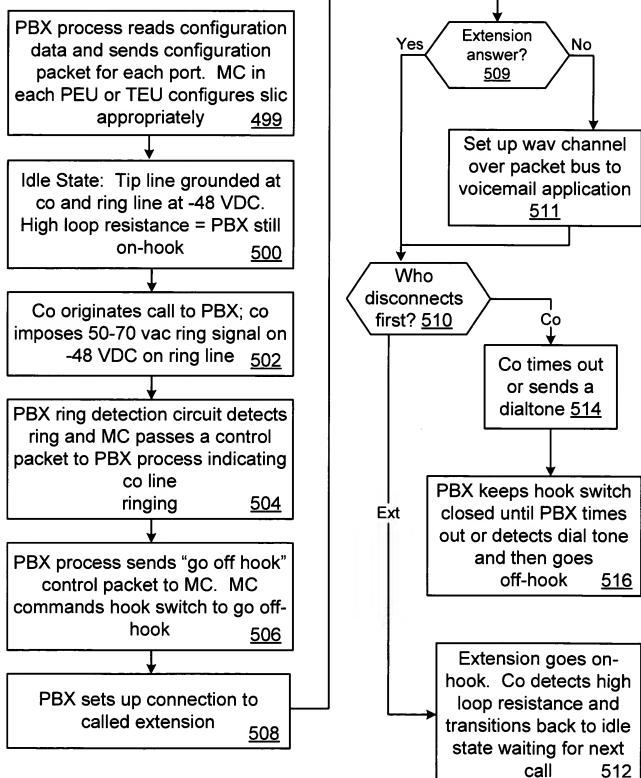


FIG. 13

Analog Loop Start Protocol Process - PBX Initiates the Call

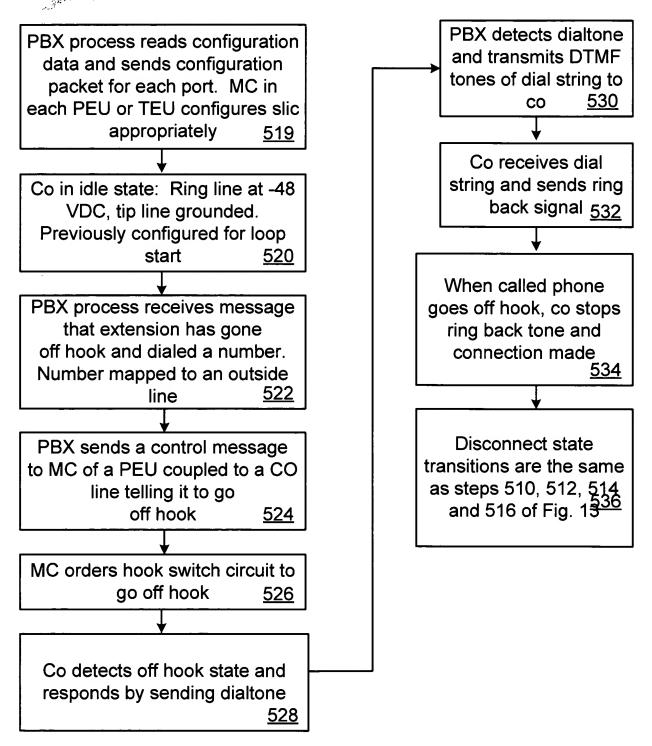
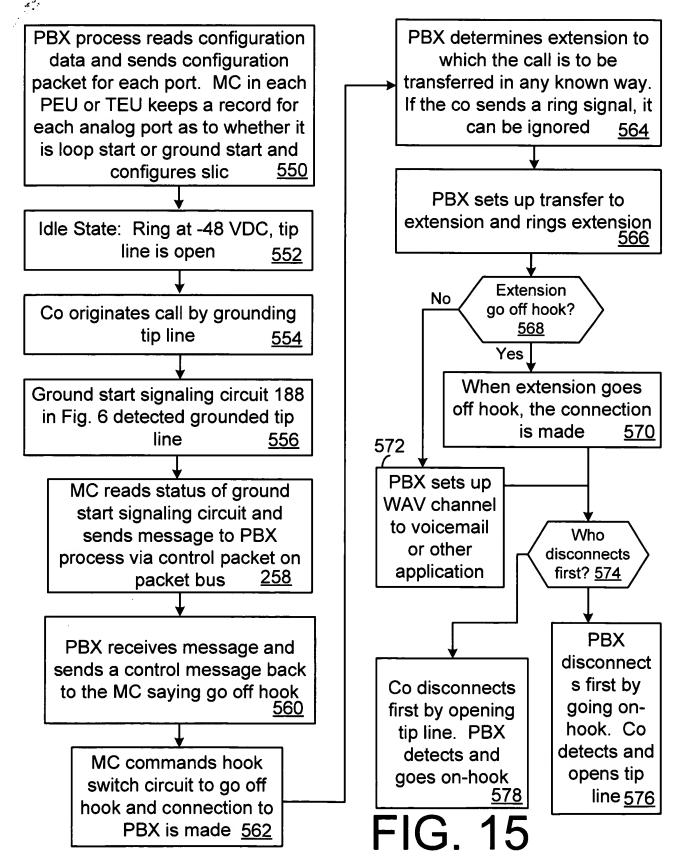


FIG. 14

Ground Start Signaling Protocol - Analog Co Line - Co Originates Call





Ground Start Signaling Protocol - Analog Co Line - PBX Originates Call

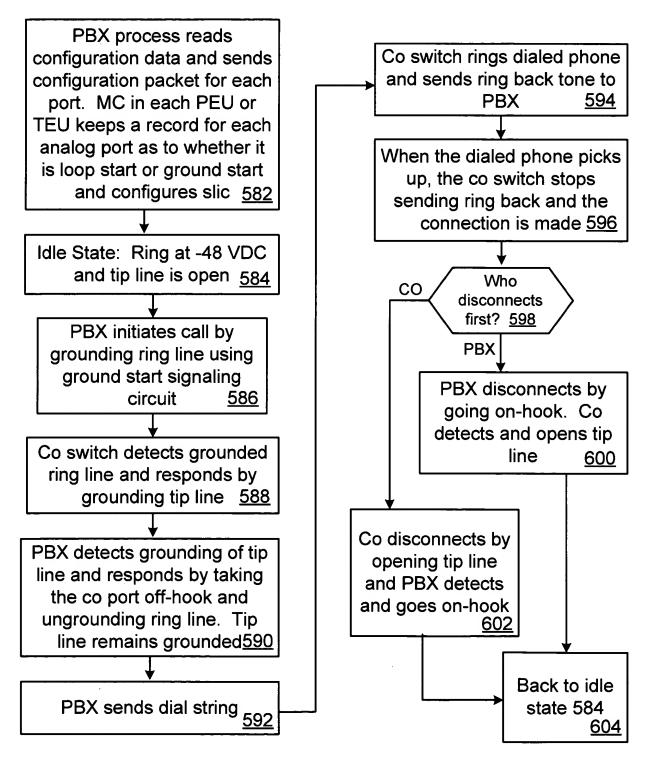


FIG. 16

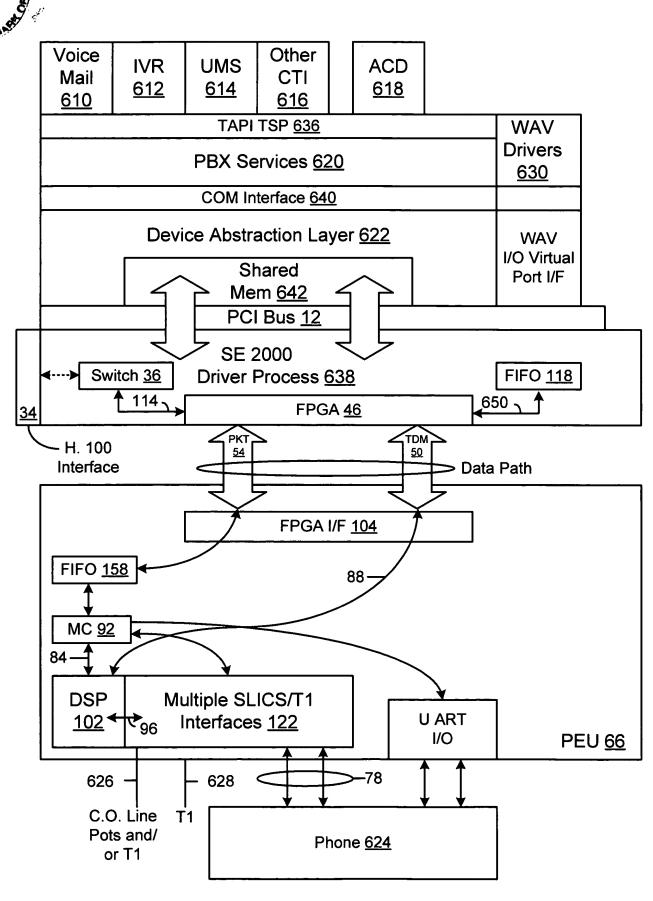


FIG. 17

